A MULTI-CHANNEL INTERIOR COMMUNICATION SYSTEM UTILIZING TIME MULTIPLEXING.

by

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THESIS

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ABSTRACT

The development of a multi-channel interior communication system utilizing a single wire as a transmission line was undertaken in this paper. The principle of time multiplexing was used incorporating the Pulse Amplitude scheme of modulation. Synchronization was accomplished by continuously transmitting a synchronization pulse from one "Master" station to all other "Slave" stations. This system permits mutually exclusive conversations between any stations concurrently. A master station and one slave station were built and tested. Using a 10-kHz sampling frequency, a frequency response of from 100 Hz to 4.8 kHz was obtained.

By using solid-state devices throughout, the size and weight of each station are minimized. This in conjunction with the need for only one connecting wire, make this system ideal for modern aircraft.

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I. INTRODUCTION

The requirement for a lightweight, multi-channel interior communication system has become apparent in modern aircraft. The Boeing 747 aircraft will incorporate a new system which meets this need, to a certain degree, in its Passenger Entertainment and Passenger Service System. This system uses time multiplexing which has eliminated approximately 25 miles of wire and saved almost 500 pounds of weight. This system is a one-way means of communication, with a single two-way feature for turning on a service light at an attendant's station. This is done remotely from any passenger's seat. The passenger entertainment part of the system can handle 15 channels of one-way audio information on one line. Reference 1 contains a complete description of this system.

A system with two-way communications on many channels, over one line, could have an immediate effect on the large military aircraft of today by reducing the weight and space required for the interior communications system.

If stations of the system could be made portable and relatively inexpensive, application aboard Naval Ships could eliminate the menial task of manning a sound-powered phone system and the inherent mistakes that occur in relaying information throughout the ship. The present 21 MC is bulky and requires large connecting cables which make it completely unsatisfactory for a portable flexible system.

II. DISCUSSION OF PROBLEM

A. BASIC PRINCIPLES

Time-division multiplexing is made possible by the fact that it is not necessary to continuously transmit a signal in order to completely reproduce the signal at the receiver. The signal may be completely specified by sampling the signal in accordance with a theorem stated by Hancock in Ref. 2.

> Theorem: If a time function contains no frequency components higher than W hertz, then the time function can be completely determined by specifying the ordinates at a series of points spaced every $\frac{1}{2W}$ seconds or less.

The recovery or reproduction of the signal may be realized by satisfying another theorem from Hancock [Ref. 2] which states:

> Theorem: If a signal whose highest frequency is W cycles has been sampled at a rate of 2W samples per second, and the samples are in the form of impulses whose areas are proportional to the magnitudes of the samples at the sampling instant, the sampled signal may be reconstructed by passing the impulse train through an ideal low-pass filter whose cut-off frequency is W cycles.

These samples may be multiplexed with samples from other sources and transmitted as a train of discrete samples. At the receiver the discrete samples are sorted and the signals reproduced in accordance with the theorem stated above.

The samples are always placed in the appropriate time position, relative to some reference, so that the receiver knows which sample belongs to which signal. If this method is to be used in an intercom system, the different stations must know the correct time to transmit and the correct time to receive. By wiring all stations to continuously either receive or transmit during a particular time interval, with the ability to select the right time to transmit or receive during any other station's time interval, a two way-communication may be held in the time interval assigned to the station being called.

For example a three-station time-multiplex system is presented in Fig. 1. Station three is to call station one by transmitting into station one's time interval. If at the same time that this selection is made station-three receiver is set to receive the station-one time slot, stations one and three transmit and receive on station-one time interval. If at the same time station two decided to call one and selected station one, he would receive one and three's communication and could talk to both. Had he elected to call station three he could hear the conversation from station three but could not hear station one. Station three can now hear both one and two.

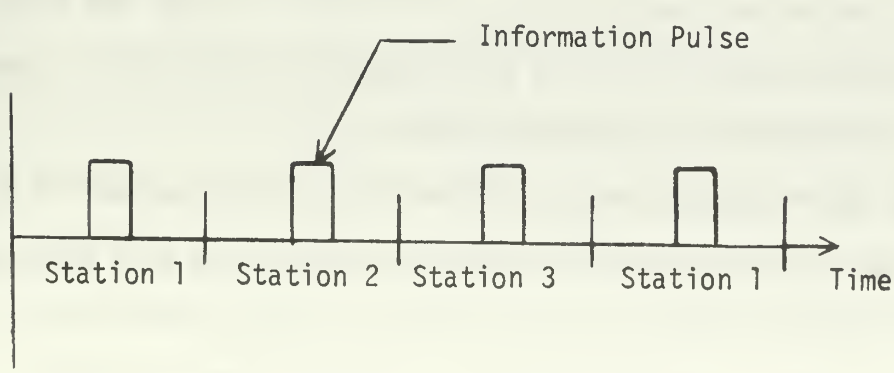

Figure 1

B. ALTERNATE PULSE-MODULATION SCHEMES

In transmitting the time-multiplexed samples the samples may take several forms. These are briefly described below so that the basic differences may be compared.

1) Pulse-Code Modulation Changes the magnitude of the sample to a digital presentation before transmission as a group of pulses. This group of digital pulses is received and converted back to the analog value of the sample.

2) Pulse-Amplitude Modulation transmits a pulse having an amplitude which is directly proportional to the amplitude of the sample.

3) Pulse-Position Modulation transmits a pulse of constant amplitude and width. The pulse is displaced from a reference time in proportion to the amplitude of the sample.

4) Pulse-Duration Modulation transmits a pulse of constant amplitude whose width is varied proportionally with the amplitude of the sample.

In an intercom system there will be varying delay times according to the station separation. The pulse-position scheme was therefore discarded as unfeasible. Also because of these delays, the guard space between pulses must be as large as possible to prevent overlap of pulses from different stations. The pulse-duration method with varying widths would therefore be unsuitable for this application. The pulse-code technique is complex but could be used. The desire to keep the cost as small as possible was considered more important than the noise immunity that is available with pulse-code modulation.

The pulse-amplitude scheme was selected as the best method for this application, because the time delays least affect it and also because of the simplicity of its circuitry.

C. TRANSMISSION AND SYNCHRONIZATION

When a system is to be time multiplexed the immediate problem is synchronization. To use a synchronization pulse requires a time interval in which audio information could be transmitted. Since synchronization requires only one discrete frequency, it seems feasible that the synchronization could be accomplished by transmitting a single sine wave and synchronizing on the zero-crossing times.

The existance of "Wireless Intercoms" in which the audio frequency is shifted to a carrier frequency, and transmitted on normal power wiring gave promise to this approach since the frequencies could be widely separated. A master station could be made from one of these existing intercoms by transmitting a sine wave at the sampling frequency and all other stations could detect this frequency and synchronize on it. The audio information could be multiplexed and stepped up in frequency, as was being done previously.

Reference 3 describes the Heathkit Model GD51A which uses an Amplitude-Modulated carrier and a diode detector so that a low-pass filter is in the circuitry. The basis for a Pulse-Amplitude Modulated system existed that would require only a power receptical to plut into. All that was required was a synchronization system and logic to control the detection of the appropriate pulses. The proposed block diagram for this circuit is in Fig. 2.

The Master Station continuously transmits a 10-kHz sine wave for the sampling frequency which would give a theoretical maximum audio frequency of 5 kHz. All stations would synchronize their multiplexers with this sine wave. The decoder would actuate a listen circuit at the appropriate number for each station. To call another station, that station's number would be channeled through a talk circuit to transmit at the appropriate time.

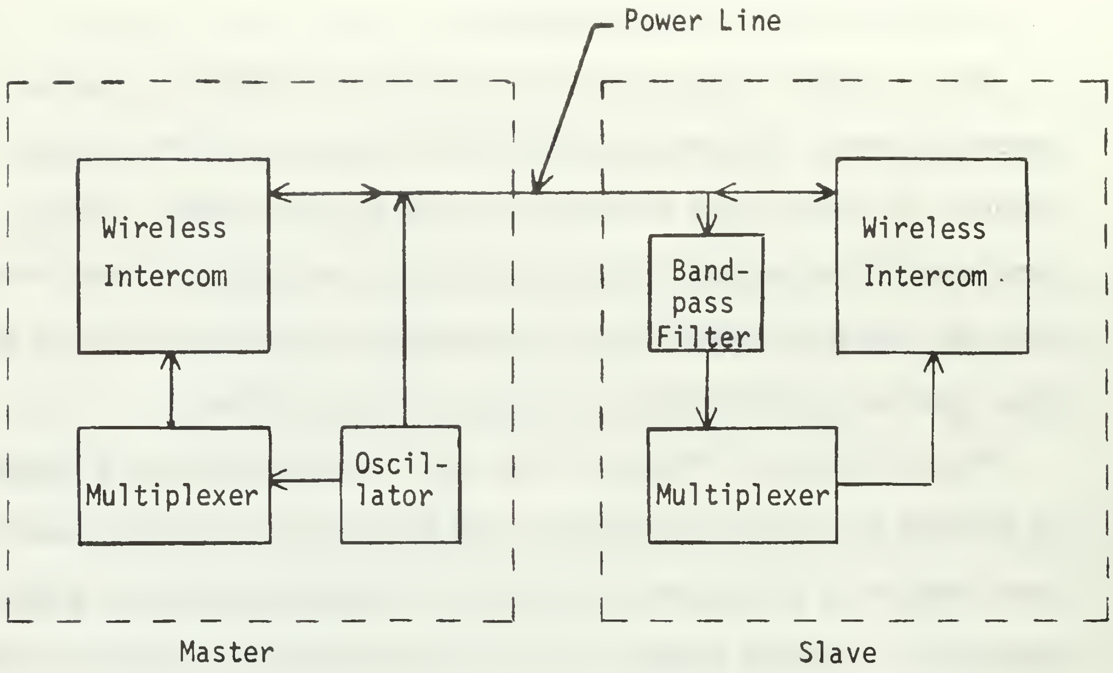

Figure 2

This approach for synchronization was abandoned after determining that electrical noise on power lines prevented synchronization to the accuracy required for multiplexing.

The idea of a wireless intercom was abandoned[1] in favor of a one-wire intercom using a synchronization pulse and no frequency stepping, as illustrated in Fig. 3. In this system the master station continually transmits a synchronization pulse. The multiplexer is synchronized to the end of this pulse.

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[1] The Wireless Intercom could be utilized by stepping up the frequency for transmission and detecting the information as explained in the next section. This solution may be realizable but the bandwidth of frequencies would be quite large and the filtering that exists on power lines of aircraft and ships would become a problem.

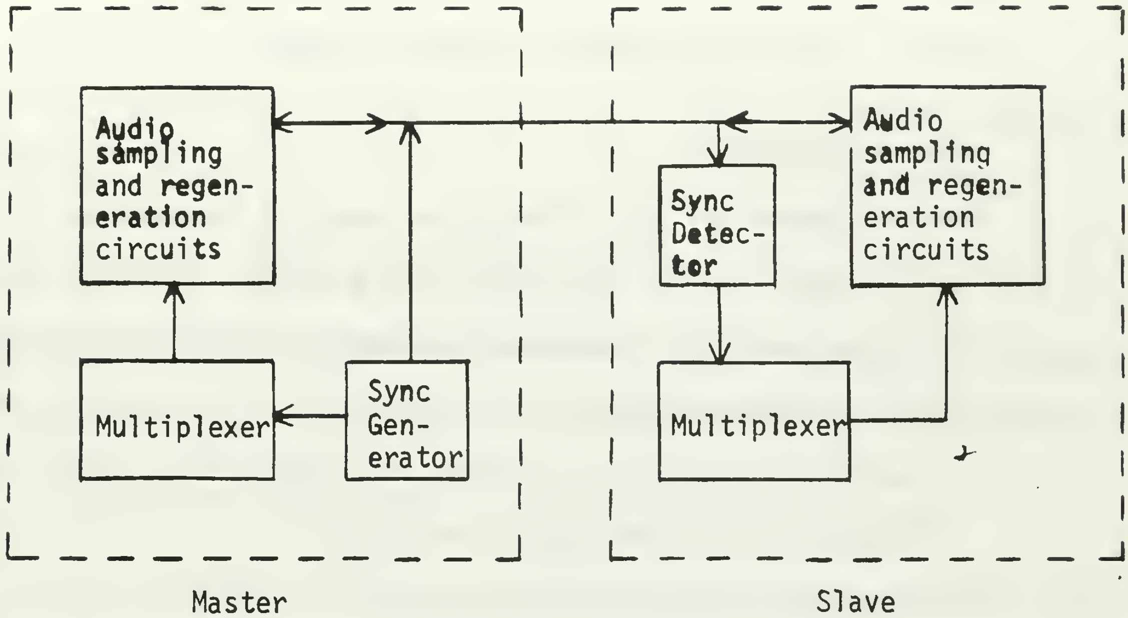

Figure 3

This is the system which was developed and tested in detail. The results are presented in the remainder of this paper.

Two methods for detecting a synchronization pulse were considered. Reference A utilized a pulse of the same width as information pulses but of a larger amplitude. A Schmitt Trigger was then incorporated to detect the synchronization pulse. An alternate method is to use a pulse with a larger width than the information pulses. Detection is accomplished by measuring the width of all pulses. This latter method was adopted primarily on the basis of power required, since the synchronization is required continually. The amplitude of the pulse is of no consequence, therefore less power is required than that where the amplitude must always be higher than the highest possible information pulse.

III. REALIZATION OF A PAM INTERCOM

A. GENERAL DESCRIPTION

The development of this intercom was based on the general availability of decade counters and one-in-ten decoders. The principles of this system could be expanded to accommodate more channels with the limiting factors being:

1) Maximum separation of stations will determine the guard space required due to the delays involved.
2) The width of the information pulses.
3) The desired highest frequency to be transmitted (sampling rate).

This system has 8 channels with one master station. All other stations are called slave stations since they depend on the master for synchronization. The slave stations are numbered 1 through 8, and theoretically any number of stations may be on a channel. Practically, the number is limited by the transmitted power. The addition of a line driver stage of sufficient power to drive the added stations would be a simple matter.

The synchronization pulse utilized was twice the width of an information pulse for ease in detecting.

A sampling frequency of 10 kHz was chosen as sufficiently high for voice intercom purposes.

The guard space between pulses was chosen equal to the information pulse width for ease in calculations and drawing waveforms. This is easily adjusted. The width of an information pulse will be referred to throughout the rest of this paper as a time interval. With two time

intervals for the synchronization pulse, 8 intervals for information and 9 intervals as guard spaces, 19 time intervals are required in one 10 kHz period, as shown in Fig. 4(a). Each time interval therefore must be:

$$\frac{1}{10^4 \text{ Hz} \times 19} = 5.26 \ \mu \text{ sec}$$

Absolute maximum separation between the master station and the most remote station, using the speed of light as propagation speed, will be:

$$2 \text{ intervals} \times \frac{5.26 \ \mu \text{ sec}}{\text{interval}} \times 3 \times 10^8 \ \frac{\text{meters}}{\text{sec}} = 1578 \text{ meters}$$

This separation allows cross talk to the next higher channel number; for no cross talk this value is halved (789 meters).

From the information pulse train the various wave forms for the block diagram may be determined.

The clock is a free-running multivibrator which has a period of 2 time intervals. The master must have the capability of generating a 2-time-interval pulse every 10^{-4} seconds. With provisions for a reset number and a number for transmitting the synchronization pulse in the master, a decade counter is left with 8 free numbers for information channels as illustrated in Fig. 4(b). The clock drives the counter, and the counter counts only on either high or low values; therefore to obtain the required 19 time intervals the clock must be stopped for one interval to get an even number of clock cycles between sample times. The clock waveform required is shown in Fig. 4(c).

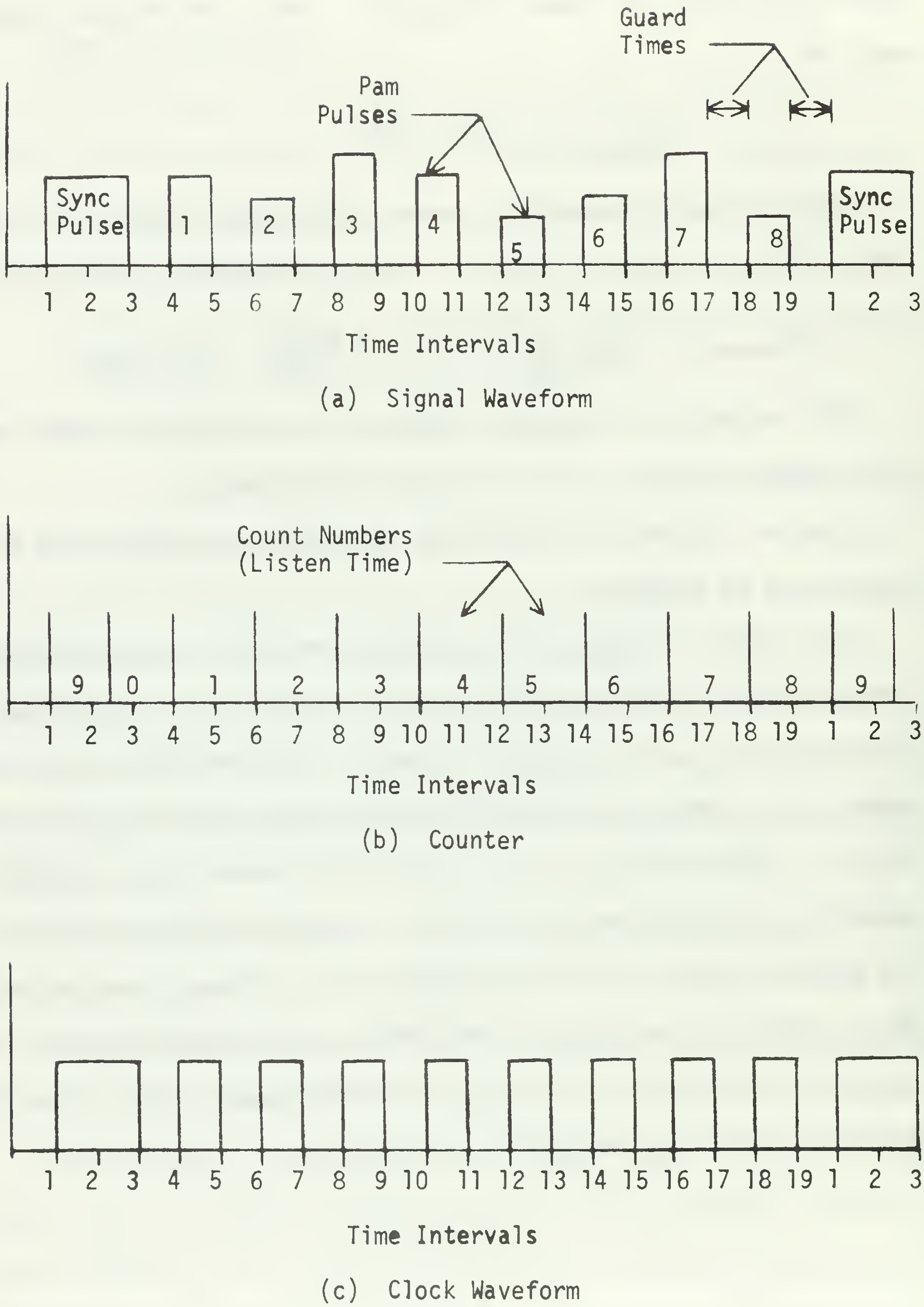
Guard Times
Pam Pulses
Sync Pulse
1 2 3 4 5 6 7 8
Sync Pulse
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 1 2 3
Time Intervals
(a) Signal Waveform
Count Numbers (Listen Time)
9 0 1 2 3 4 5 6 7 8 9
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 1 2 3
Time Intervals
(b) Counter
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 1 2 3
Time Intervals
(c) Clock Waveform

Figure 4

B. LOGIC CIRCUITS

1. Clock Circuit

For the clock circuit a free-running multivibrator is used. This was formed by using two Fairchild 9601 retriggerable monostable multivibrators. These were used because of their high accuracy and the logic functions built into the integrated circuits. The period of these monostable multivibrators may be controlled by external resistances and capacitors to a very high degree of accuracy. References 5 and 6 contain the characteristics and varied applications of these integrated circuits. The logic of the input allows the clock to be stopped through either OR or AND gates which gives considerable flexibility in the selection of other logic elements. The packaging of two of these on one integrated circuit chip in the near future[2] will make the clock function even more compact.

2. Synchronization Generator

For the synchronization pulse generator another 9601 multivibrator was used for the reasons given above. The 9601 is triggered by the decoder going to count 9. The output of the synchronization generator stops the clock, resets the decoder to count zero, and transmits the synchronization pulse to the other stations. At the end of the synchronization pulse the clock starts and after one time interval the decoder goes to count one.

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[2] This information was obtained from Michael English of Fairchild Semiconductor, Applications Laboratory, during a discussion of this project.

3. Synchronization Detector

Again the 9601 was utilized, this time to determine the width of arriving pulses. As a pulse arrives the one shot is triggered, timing for one and a half time intervals. If a pulse is present at the end of this time the clock stops and the counter is reset to count zero. The presence of the pulse is determined by putting the incoming pulses and the output of the 9601 into a NAND gate; if both are true a false indication appears at the output. For the NAND gate a Fairchild Transistor-Transistor Micrologic 9002 was utilized because of its compatability with other Integrated Circuits in the logic and the number of NAND gates in one package. References 7 and 8 give the complete electrical and physical characteristics of the 9002.

4. Multiplexer

The Fairchild Counter Micrologic Integrated Circuits Decade Counter 9958 and Counting Micrologic Integrated Circuit Decimal Decoder/ Driver 9960, one-in-ten decoder, were used as the multiplexer. The 9958 counter was chosen because of its compatability with the other logic elements and the 9960 for the same reason. The use of the 9960 had one serious drawback. Since it was designed primarily as a driver for gas-filled cold-cathode indicator tubes, the utilization of the decoder function required a return to the supply voltage through a 10 KΩ resistor for each output. References 9 and 10 contain the electrical and logical characteristics of 9958 and 9960.

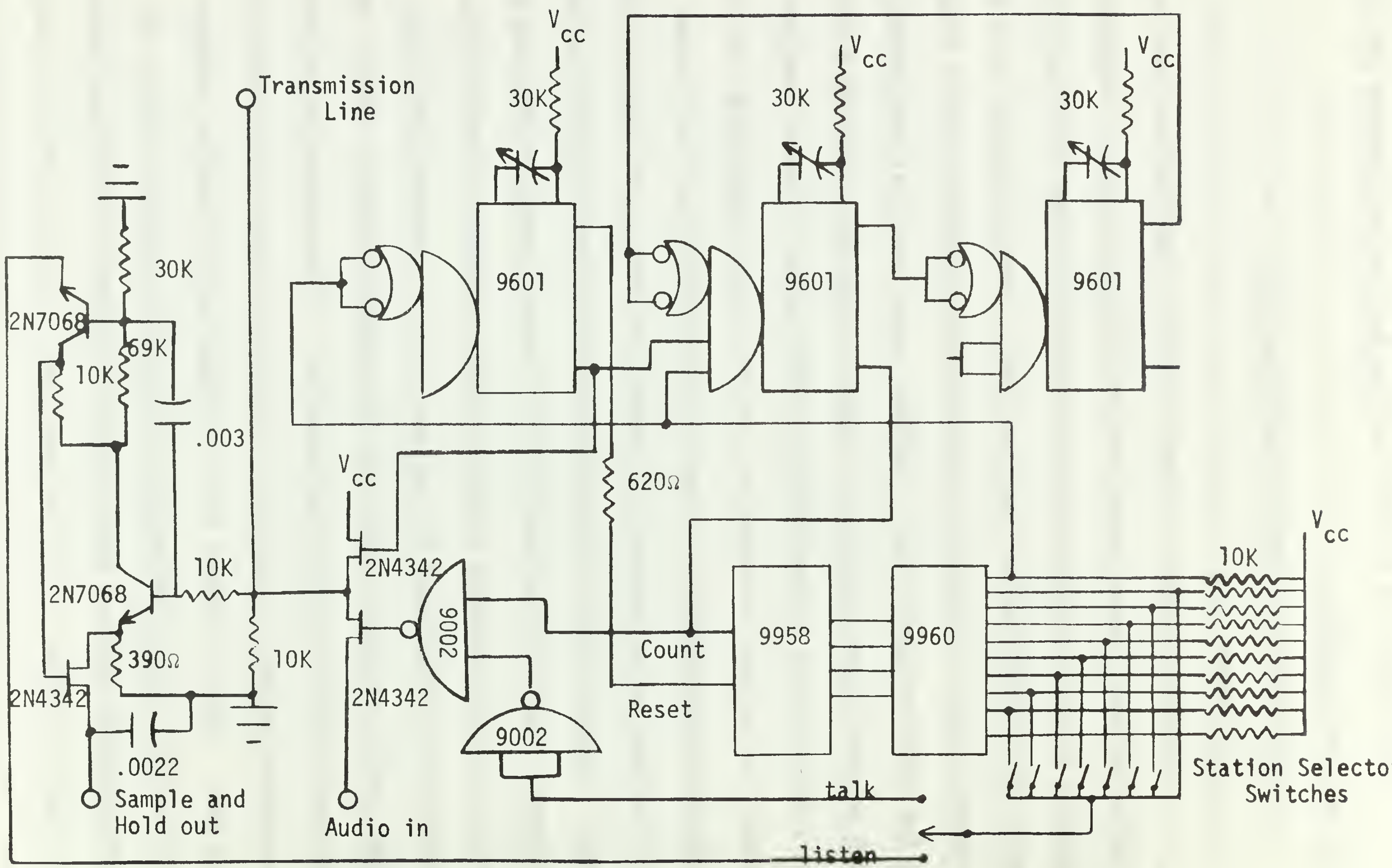
The 9958 counts on a positive-going pulse and resets to zero on a positive pulse. The reset has a low input resistance (300Ω) so it was necessary to use a series resistance of 620Ω to present the proper load to the driving element.

The outputs of the 9960 are low for the "on" numbers and high for the "off" numbers.

5. Description of Combined Logic Operations

(a) Master: (See Fig. 5)

As soon as supply voltage (5 volts) is applied the clock begins running and the counter counts the positive-going pulses. When count number 9 is on, the return from the decoder stops the clock by its low value into the AND gate of the first 9601. The count number 9 is also returned to the OR gate of the synchronization generator which triggers on the negative-going slope. The high output from this 9601 resets the 9958 counter to zero and the low output is applied to the AND gate of the clock with the 9 output from the 9960 decoder. By the time the counter and decoder are reset to zero the input to the clock from the synchronization generator is low, thereby keeping the clock stopped. The low output also turns on a P-channel, 5-V pinch-off junction field-effect transistor which transmits the synchronization pulse to the other stations. The Fairchild 2N 4342 Field Effect Transistor (FET) was chosen for this function because of its favorable parameters. When the two-interval time has elapsed the outputs from the synchronization generator shift, the clock starts and the counter is able to count. After one time interval the clock goes positive. This positive pulse shifts the counter decoder to count one; it also puts a positive pulse on a 9002 NAND gate. Then if channel one is selected on the selector switches on the output of the 9960 decoder and the talk switch is closed, the inverted output (through another 9002) activates the NAND gate causing the output to go low. This low value turns on another 2N4342 FET which closes the circuit between the audio out and the transmission line. As



Transmission
Line
Vcc
30K
9601
Vcc
30K
9601
Vcc
30K
9601
30K
2N7068
69K
10K
.003
Vcc
620Ω
Vcc
10K
2N4342
10K
2N7068
9002
9958
9960
390Ω
10K
Count
2N4342
2N4342
Reset
9002
.0022
Station Selector
Switches
Sample and
Hold out
Audio in
talk
listen

Figure 5

soon as the clock goes negative (after one time interval) the NAND gate goes high opening the audio-transmission circuit. This has the effect of multiplexing the system. If the talk-listen switch was in the listen position the emitter of a high-speed switching transistor 2N 706B is low for two time intervals. If during this time there is a pulse arriving from the line the transistor turns on. This switches another 2N4342 FET on, permitting a capacitor to be charged to a level proportional to the amplitude of the signal pulse. When either the line pulse is completed or the listen time interval is over, the 2N706B transistor switches off and the 2N4342 opens the circuit to the capacitor forming a sample and hold network. The samples are allowed only if the desired listen channel and line pulse are present. This forms the de-multiplexing function of the circuit.

(b) Slave: (see Fig. 6)

The only difference in the slave and the master circuit is that the synchronization generator is replaced by a synchronization detector. The incoming pulses all trigger the synchronization detector and also go to one input of a 9002 NAND gate. The output from the 9601 is normally high but when triggered goes low for one and one-half time intervals; this output is applied to the other input of the NAND gate. If the pulse is present on the line after one and one-half time intervals the output of the NAND gate goes low and stops the clock. The output also triggers another NAND gate with the inputs wired together forming an inverter. This drives the reset of the 9958 counter, resetting to zero. As in the master, the synchronization pulse keeps the clock stopped until the end of the pulse drives the output from the first NAND gate high. This starts the clock and permits the counter to accept

counts by forcing the reset terminal low. The 9 count from the decoder is routed to the input AND gate of the first 9601 so that the clock will always be stopped long enough for it to be in the correct state to be started again by the synchronization pulse to the first 9601. (If the second 9601 were in its timing cycle the first 9601 could not be triggered.)

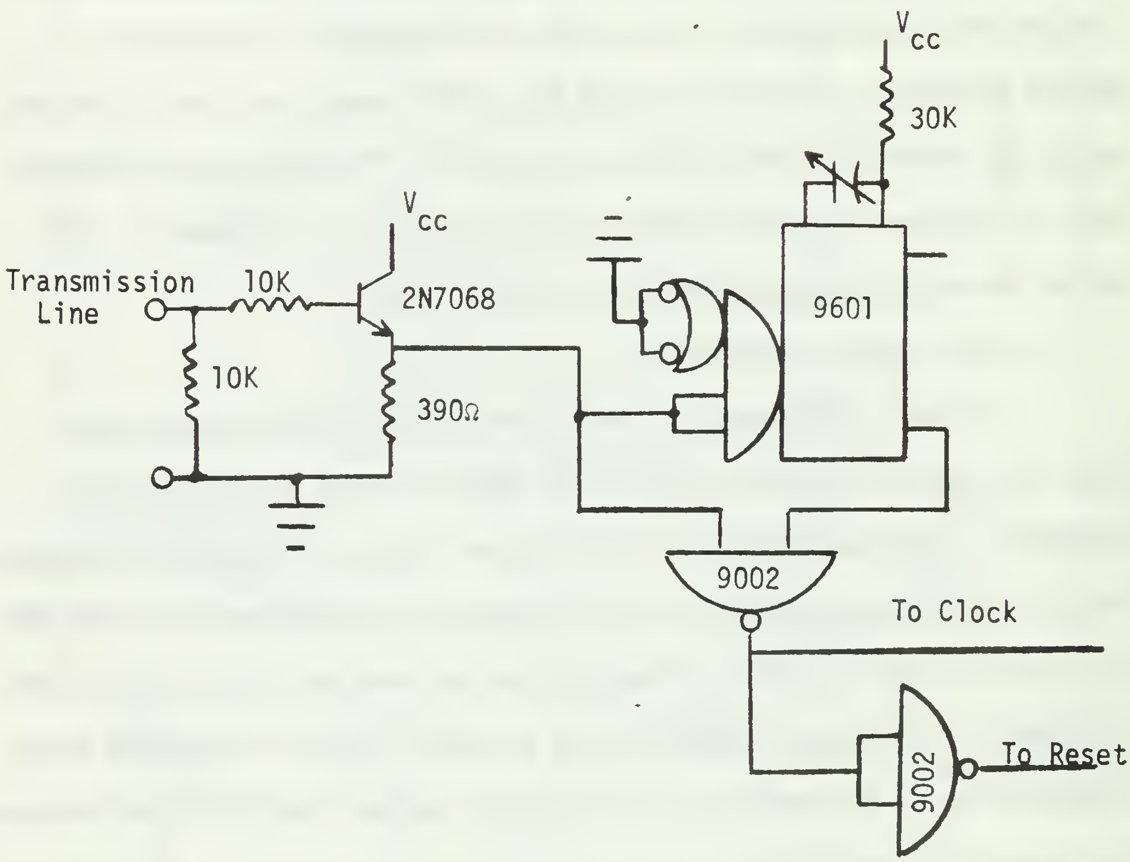


Figure 6.

A 2N 706B transistor wired as an emitter follower is used to drive the logic, the sample-and-hold circuit, and one input to a NAND gate. The maximum low current for the 9601 is 1.6 ma and for the 9002 is 1.76 ma; the maximum low input voltage is 0.8 volts. Therefore the emitter resistor should be $\frac{0.8 \text{ volts}}{2.36 \text{ mA}} = 340\Omega$. A value of 390Ω was chosen in order to be safe and to maintain a sufficiently high input impedance.

C. AUDIO SAMPLING AND REPRODUCTION CIRCUITS

1. Sampling Circuit (See Fig. 7)

As a sampling circuit another Fairchild Field-Effect transistor was used as a chopper. The FET was used as a series element as explained in Ref. 11. The 2N4342 P-channel junction FET was used because of its pinch-off voltage of under 5 volts which permitted the logic to drive the chopper directly. The audio signal into the source of the FET has a D.C. value of 5 volts so that the sampled pulses would be unidirectional with a value from 1 to 9 volts. This one-volt D.C. value was required to trigger the sample-and-hold circuit at the receiving station as explained later. The drain of the FET was attached to a 10-KΩ resistor across the transmission line. The 2N4342 can power 10 stations before its maximum current rating is exceeded by the loading of other stations.

The proposed method of generating the required audio signal was by the use of a 45-ohm, one-watt combination speaker-microphone. This would drive a Fairchild Monolithic Operational amplifier 741. This operational amplifier may be modified by external resistors to obtain gains of from 1 to 500 with a minimum input impedance of 100 MΩ over the required bandwidth. This flexibility allows the 741 to be used in the receiving circuit as explained later. The characteristics and versatility of the operational amplifier are given in Refs. 12 and 13.

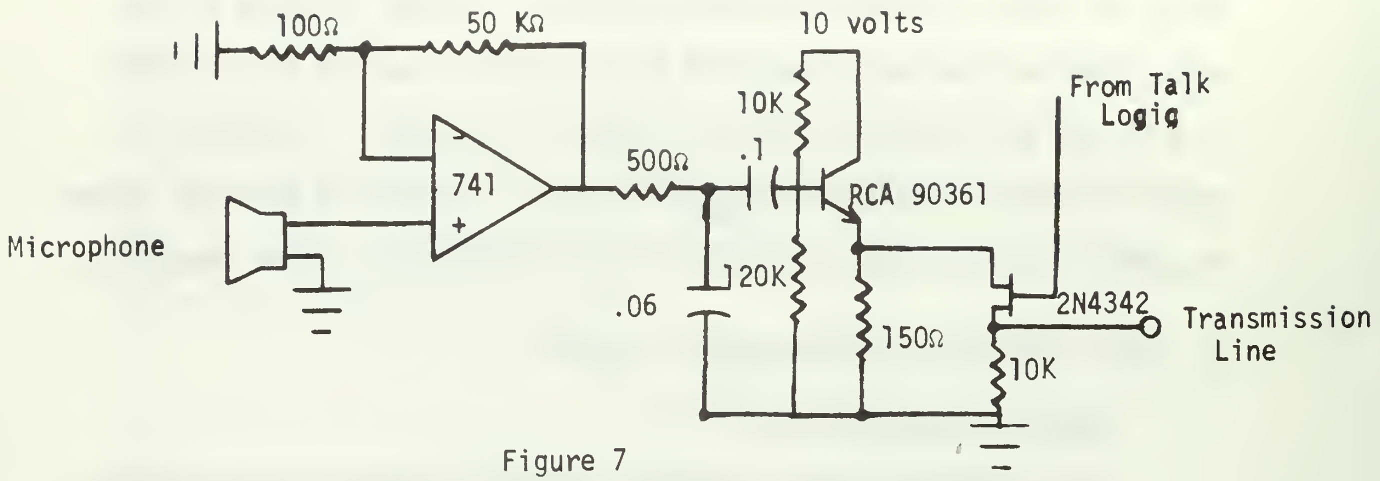


Figure 7

The operational amplifier gives the A.C. voltage required to drive the Sampling Gate. To obtain the D.C. value this was A.C.-coupled to an emitter-follower circuit. The RCA 40361 Transistor was used for this because of its very high h_{FE} of 300 and its 5-watt power dissipation. The importance of these characteristics is explained in the next section.

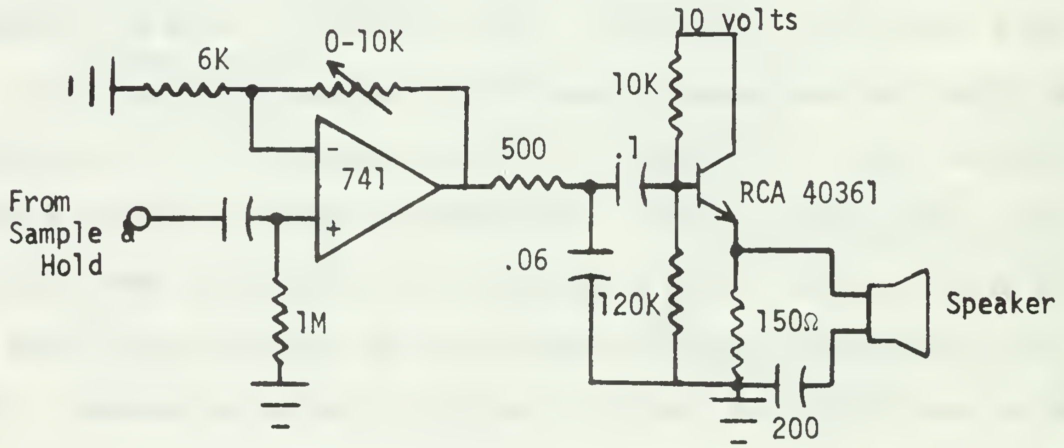


Figure 8

2. Signal-Reproduction Circuit (See Fig. 8)

As signals arrive at a station the 2N7606B transistor is driven into saturation if the listen logic is correct, as explained in section III.B.5.; the 2N4342 FET is driven on and the capacitor at the drain is

charged to a voltage proportional to that on the line. The minimum line voltage that consistently drives the 2N706B into saturation is one volt (this determines the minimum allowable pulse amplitude which must be transmitted). As soon as the amplitude of the pulse goes low the FET turns off; creating a sample-and-hold system, if the capacitor is made to hold its charge. This was accomplished by using the Operational Amplifier 741 again as a high-impedance input. Volume control was possible by varying the feedback resistor on the 741. Since the sample-and-hold output had a D.C. value, and only the amplitude of the audio needs controlling, a coupling capacitor was necessary. A 1-MΩ resistor was used to provide a return to ground on the input of the 741. A low-pass filter was used at the output of the 741 to eliminate most of the 10-kHz sampling frequency which remained in the signal. To provide the power required to drive the speaker it was proposed that an emitter follower be used. The RCA 40361 had a high h_{FE}; therefore it would provide the impedance at the input to prevent interference with the proper operation of the low-pass filter. It also has power dissipation capabilities to drive the one-watt speaker. In the desire to keep the number of elements and cost as low as possible, the selection of the above elements permitted their use in both the talk and listen positions without redundancy. By using two sets of external resistors on the 741 the required gain values may be obtained for both the talk and listen functions, by means of the talk-listen switch. The only possible problem was the extra loading from the output to ground by the unused set of resistors. This actually was insignificant, the lowest possible value being 6 KΩ.

The combined functions are illustrated in Fig. 9.

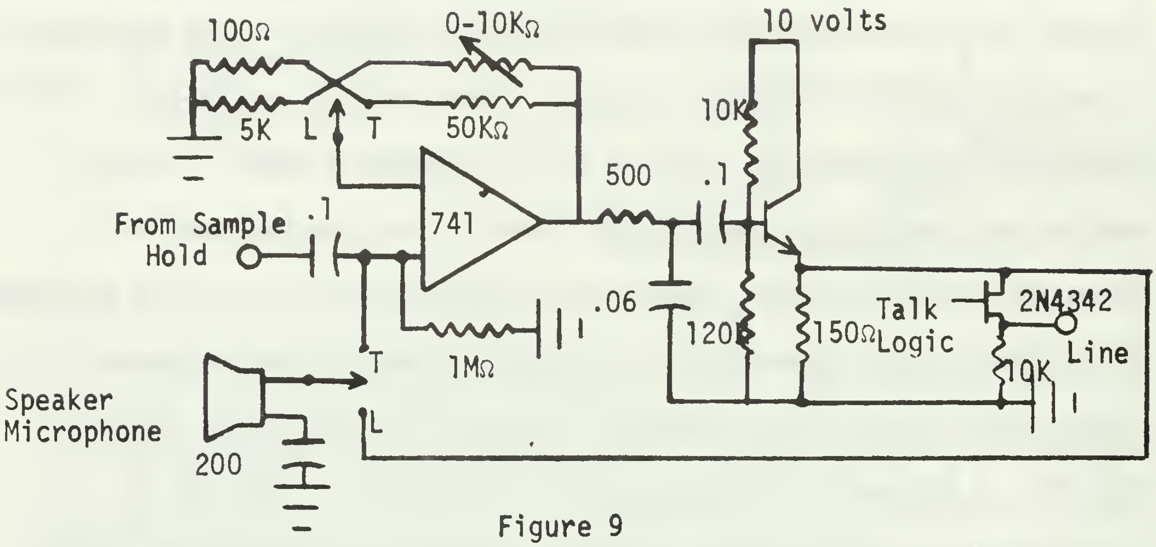


Figure 9

IV. CIRCUIT PERFORMANCE

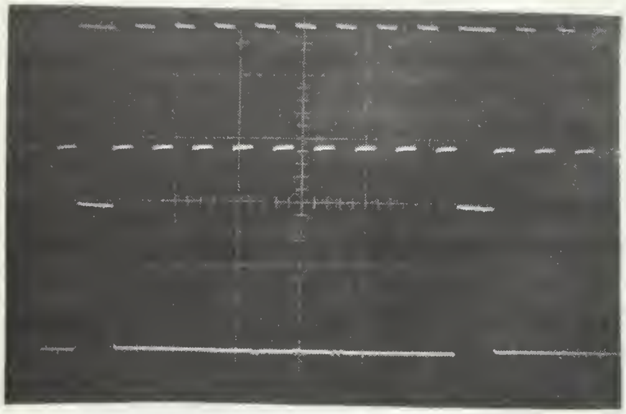
A. BASIC CONSIDERATIONS

To evaluate circuit performance a Master station and one Slave station were built. The stations were connected by a four-foot laboratory test lead. No work was done on allowable station separation or optimum transmission-wire characteristics. Each station was provided with a 5-volt supply for driving the logic, and $\pm$ 10 volt supplies for the audio. A dual-trace oscilloscope was helpful in comparing different aspects of the circuit. All waveforms used in the audio evaluations were connected to the operational amplifier input.

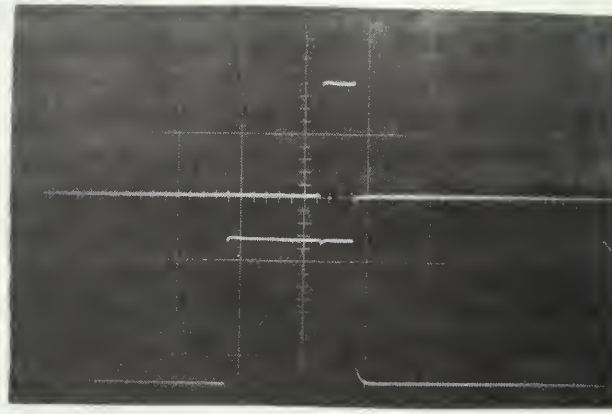
B. LOGIC PERFORMANCE

The Master logic to form the synchronization pulse and the correct count sequence may best be determined by looking at the Master station clock. This is presented in the top of picture one; the bottom section is a picture of the synchronization pulses as they appeared on the transmission line. The amplitudes of the clock pulses were about 4 volts and the synchronization pulses about 4.5 volts. The separation between the synchronization pulses was 100 μ sec; it can be seen that the transition times were very good.

Picture 2, bottom section, shows the synchronization pulse, greatly expanded, on the transmission line together with the slave synchronization detector output. As can be seen this output is exactly synchronized with the synchronization pulse. Picture 3 has the synchronization pulse, from the same location as the first two pictures, on the bottom; the slave clock is on the top. Upon close observation of the picture it may be

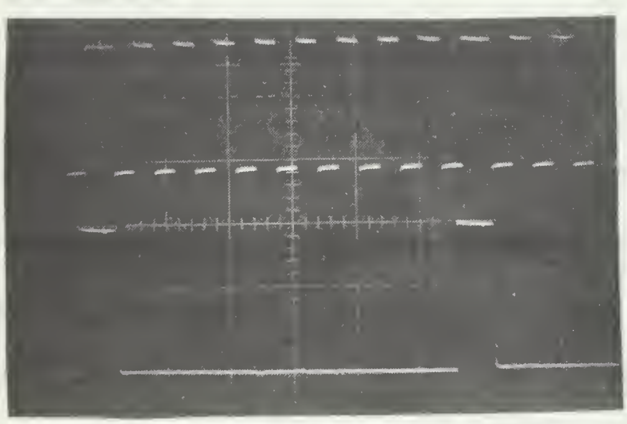


1.



2.

seen that the last low pulse of the clock, during the entire sample shown, overlaps the synchronization pulse by approximately 2 μ sec. In other words the slave clock is running slow. This is easily adjusted by the trimmer capacitors on the clock, but illustrates here that the clocks need not be of extremely high accuracy. The clock could be almost three times intervals slow (about 15.7 μ sec) in each sample time (100 μ sec). The adverse effects are cross talk on the higher stations and channel eight completely blocked (The sample-and- hold



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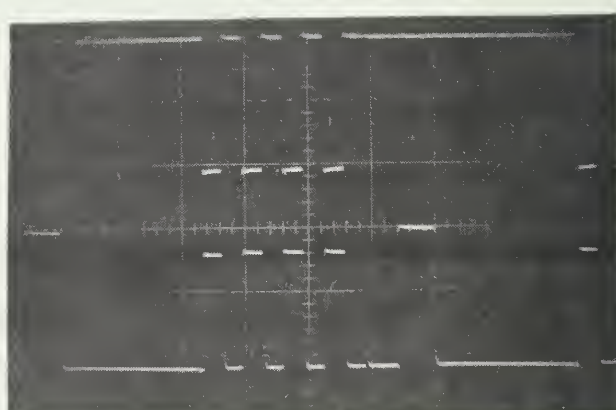
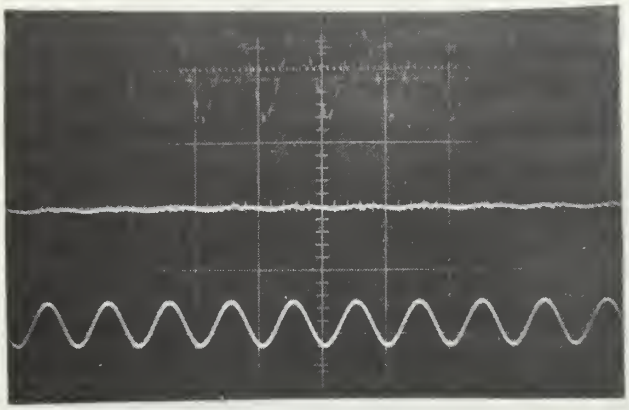


4.

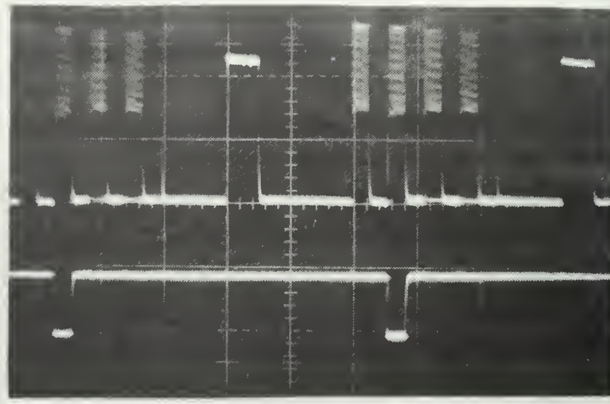
circuit of channel eight would receive the synchronization pulse each time). A variation of one time interval has no effect on the performance in the listen position, but may cause cross talk to the next lower channel than the one called. This would depend on the separation of the stations and the amount the clock is slow. If the clock were running fast, the same effects would be observed as above, with the only difference being that the talk-listen situations are reversed. The accuracy of the 9601 prevents these extreme situations from occurring if they are initially adjusted correctly.

The top section of picture 4 is the gate of the sampling FET with channels 4, 5, 6 and 7 selected. A D.C. voltage is applied to the source through the emitter-follower transistor. This voltage appears on the transmission line, with the synchronization pulses, as 4 evenly spaced pulses, as shown in the bottom section of photograph 4. A close examination of the photograph reveals a "spike" after the last information pulse one time interval away. This always appears when at least one channel is on, always one time interval from the last information pulse. This was probably caused by the varying delay times of the pulses going into the NAND gate that controls the sampling FET. The clock pulse is fed directly into the NAND gate and it also controls the 9958 counter which controls the 9960 decoder, the output of which goes through an inverter into the same NAND gate. Since delay times were given for the counter and inverter (maximum of 300 μ sec total), and none was given for the decoder, it was blamed as the element that was too slow.

With the exception of the extraneous "spike" the logic performed as was expected. The power required for the logic was 0.45 watts for the slave and 0.5 watts for the master.



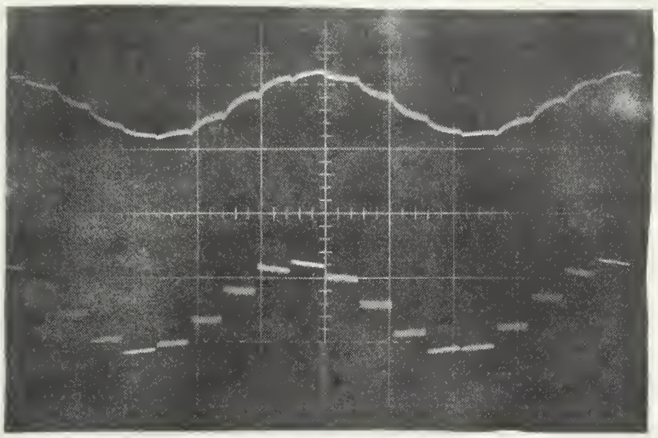
5.



6.

C. AUDIO

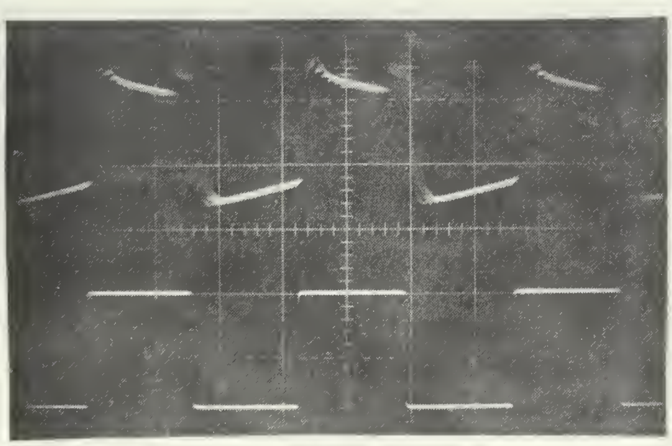
Although no bandwidth data was taken, frequencies from 100 Hz to 4.8 kHz were put through the system successfully. Success was determined by the ability to hear the tone without harmonic distortion on the output speaker. Photographs of the output waveforms of the higher frequencies were impossible due to their distortion in combination with the oscilloscope trigger. A 1-kHz sine wave was photographed at various points in the system. The bottom section of photograph 5 is the input. The top is the line waveform with the oscilloscope triggering on the input sine wave. The transmitting station (Slave station) was on 4 channels, 3, 4, 5, and 6, so that the sine wave appears as a dashed line. The synchronization pulse may be seen as a straight line of dots just above the center of the transmitted sine wave. The bright line through the center of the picture is the ground potential of the line. Photograph 6, top section, is the line signal with the oscilloscope triggering from the output of the decoder's channel one. The Master station was wired as channel 4 in this instance and the lower part of the photograph shows the



7.

voltage applied to the sample-and-hold gate. The voltage levels are from about +5 volts to 0 volts. Photograph 7, bottom section, shows the output of the sample-and-hold circuit. The upper section shows the 1-kHz signal after it has passed through the 741 operational amplifier and the low-pass filter.

The final photograph is of the responses to a 300-Hz square wave. In the bottom section of picture 8 is the input to the 741 and at the



8.

top is the output of the filter at the other station. The distortion in conjunction with the trigger makes accurate information impossible, but a crude approximation of the rise and fall times is about 0.2 μ sec which gives a cutoff frequency of about 5 kHz.

The audio part of the circuit performed sufficiently well as far as the requirement for a voice intercom is concerned.

V. CONCLUSIONS

This paper has demonstrated the feasibility of a multi-channel one-wire interior communications system. By using integrated circuits throughout the system, the weight of each station would be primarily due to the speaker and the case. Although no detailed cost analysis was done, it has been estimated that in mass production each station could be produced for around $100.00. These characteristics make use of these in modern aircraft not only feasible but desirable. On new-construction ships this system could probably compare favorably in price with the presently used 21 MC when all the costs of cable and installation are included.

Details which warrant more work are first the location of a better counter-decoder combination. The delay time through these elements is excessive and the requirement that the decoder have external resistors is not practical. A power supply with the required voltages should be built. Investigation into the type of transmission lines best suited for this type application should be pursued. This could also be extended to a study of the existing sound-powered phone lines aboard ships to determine if this wiring could be used as transmission lines for this system. An evaluation of the maximum separation of stations and the system's performance under these circumstances should be made.

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13. ABSTRACT

The development of a multi-channel interior communication system utilizing a single wire as a transmission line was undertaken in this paper. The principle of time multiplexing was used incorporating the Pulse Amplitude scheme of modulation. Synchronization was accomplished by continuously transmitting a synchronization pulse from one "Master" station to all other "Slave" stations. This system permits mutually exclusive conversations between any stations concurrently. A master station and one slave station were built and tested. Using a 10-kHz sampling frequency, a frequency response of from 100 Hz to 4.8 kHz was obtained.

By using solid-state devices throughout, the size and weight of each station are minimized. This in conjunction with the need for only one connecting wire, make this system ideal for modern aircraft.

14 KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Time Division Multiplexing						
Synchronization (digital)						
Pulse Amplitude Modulation						
Multivibrators						
Digital Logic						
Chopper Circuits						